Kindly replace the paragraph beginning on page 3, line 24 to page 4, line 9 with the following amended paragraph:

The upper part of ~~fig.~~ Fig. 1 shows a system 1, which is suited for suppressing audio distortion in a desired signal. The system as shown has a loudspeaker 2 and a microphone array 3 comprising two microphones 3-1, 3-2. An audio output signal on output 4 is reproduced by the loudspeaker 2. A near end source (not shown) generates desired speech, which is received by the array 3 as a desired speech signal. In addition the microphone array 3 senses [[-]] (as clarified in connection with Fig. 2) [[-]] as part of different kinds of distortions apart from noise, (i) a direct signal from the loudspeaker 2 to the array 3, (ii) echoes in the form of early [[-]] (first part) [[-]] reflections and (iii) after some exponential decay, later [[-]] (second part) [[-]] reflections in the form of so called reverberation shown as a reverberating tail of a typical room impulse response as a function of time. Each microphone 3-1, 3-2 may have its associated echo canceller g_1, and g_2 respectively coupled between the audio output 4 and the distorted desired audio sensing microphone array 3. If at all possible hardware and/or software parts of the echo cancelling means g_i (i = 1, 2 for two microphones) may be used in common in order to save costs. Each of the echo cancellers g_i simulate the path from the loudspeaker 2 to a respective microphone 3 in order to cancel the effects of at least the direct signal and the early reflections, that is the first part of the echo. The technique accomplishing that is for example known from WO 97/45995, whose disclosure is incorporated herein by reference thereto. The respective echo cancelling means may be implemented in various ways, such as with Least Mean Squares (LMS), Recursive Least Squares or Frequency Domain Adaptive Filter using Block LMS techniques.

PATENT

Appl. No. 10/596,762   Docket No. NL040037US1
Response to Office Action of December 1, 2009   Customer No. 24737

Kindly replace the paragraph on page 4, lines 13 to 30 with the following amended paragraph:

The system $\underline{1}$ has a filter arrangement 7, which may include a beamformer 7B, which is coupled through the subtractors 5 to the echo cancelling means g_i and/or to the microphone array 3. The beamformer 7B, which is included in a generally called Generalised Sidelobe Canceller, is capable of defining and controlling an audio microphone sensitivity lob or curve. Given the, in this case, two beamformer input signals on the subtractor outputs 6-1, 6-2, these signals comprise the desired audio/sound/speech signal and a reverberation signal originating from the reverberating tail (e.g., reflections in the form of reverberation (second part) as shown in Fig. 2). The beamformer 7B is capable of discriminating the reverberation signal by deriving a primary signal z including the desired signal and a reference signal x which includes the reverberation. It does this here by filtering in filters f_1 and f_2, as shown, and then summing in summing device 9-1 the filters f_i outputs (i.e., the outputs of filters f_1 and f_2) to reveal the primary signal. This way the echo cancelled microphone signals u_1 and u_2 (i.e., corresponding to the outputs of microphones 3-1 and 3-2) are added such that remaining direct signals and early reflections of the desired audio are coherently summed, which increases the ~~beamformers~~ beamformer's performance. Furthermore it does this here by filtering the echo cancelled microphone signals (6-1 and 6-2 of Fig. 1) in blocking filters b_1 and b_2 (i.e., together represented by "B" in Fig. 1) and then by summing in device 9-2 the filters' outputs to reveal a reverberation representing reference signal x. The reference signal x virtually contains no desired signal components. The filters b_i (e.g., filters b_1 and b_2), together B, are called the blocking matrix. The filters f_i and b_i carry the directional, that is the desired sources, dependent information.

Kindly replace the paragraph beginning on page 4, line 31 to page 5, line 8 with the following amended paragraph:

In the case as shown in Fig. [[3]] 1, the beamformer 7B has one delay element 8 coupled to output 10 of <u>summing</u> device 9-1 followed by a summing device 9-3. The delay element 8 provides a non causal part to the beamformers' impulse response which appeared to improve its performance. The reference signal x is fed to an adaptive filter, indicated w in Fig. 1, whose output signal is fed to an inverting input 11 of <u>summing</u> device 9-3. The filter w of the filter arrangement 7 comprises the filter coefficients which represent or contain a measure for the reverberation [[–]] <u>(see, for example, Fig. 2, labelled (second part</u>)) [[-]] distortion in the desired audio sensed by the microphone array 3. The summing device 9-3 also has a summed or beamformer output S used to adapt the filter coefficients in the adaptive filter w of the thus adaptive filter arrangement 7, such that their coefficient values represent the varying reverberation distortion. In a non adaptive embodiment the filter coefficients would be fixed to then cancel a then presumed fixed reverberation tail.

Kindly replace the paragraph on page 5, lines 12 to 20 with the following amended paragraph:

Fig. 3 shows an embodiment of a filter arrangement 7 having an array of three microphones 3-1, 3-2, 3-3<u>, each microphone having an output u_1, u_2 and u_3, respectively</u>. Essentially<u>,</u> a plurality of microphones is possible. However<u>, the</u> above outlined principles remain the same. Block matrices may be grouped into one block B. Different reference signals x_1 and x_2 may be fed to the filter 7A, here comprising generally adaptive individualised filters w_1 and w_2. ~~At wish~~ <u>As desired,</u> delay elements Δ may be divided up in front or after the filters f_1, f_2, and f_3 coupled to the respective three microphones 3. Separate delay elements Δ could be included in the respective

branches from possibly each of the microphones to summing device 9-1 to account for expected individual delays between loudspeaker 2 and microphone 3.

Kindly replace the paragraph on page 5, lines 26 to 29 with the following amended paragraph:

Although the above description of the figures is related to a filter arrangement 7 embodied by a beamformer 7B, it should be noted that the above also holds for any filter arrangement 7 which is aimed at suppressing audio distortion in general in the system 1 as elucidated in the above.

Kindly replace the paragraph beginning on page 5, line 30 to page 6, line 11 with the following amended paragraph:

Reverberation is a form of distortion, whose cancellation is strongly dependent on the spatial correlation properties of the microphone array formation and the room concerned. The spatial correlation of the sound field determines the mutual correlation between the microphone signals, which in turn is the quantity which determines the design and performance of the filter arrangement 7 and beamformer 7B. These spatial correlation properties depend on several properties such as room geometry, wall absorption, position, direction and spacing of the microphones of the array 3 in a room. Advantageously the system 1 does not require some kind of model for these properties. When however these properties and/or modelled distortion echo cancelling properties or coefficients in the system 1 change significantly then substantial adaptation thereof is required, which takes a considerable amount of time. In general this time is not always available having the consequence of audible and disturbing effects during in particular shortly after presence or after absence of especially speech. The above is of primary importance in relation to spatial correlation dependent forms of distortion, such as

reverberation. The remainder of this description will therefore be directed to cancellation of reverberation by means of the system having a filter arrangement as shown in figure 1.

Kindly replace the paragraph beginning on page 6, line 12 to page 7, line 2 with the following amended paragraph:

The lower part of ~~fig.~~ Fig. 1 shows a circuit arrangement 7' whose components are at least partly similar to [[–]] (i.e., or mirrored relative to) [[-]] ~~to~~ the echo cancelling means g and the beamformer filter 7B. It comprises the beamformer 7'B which may have parts similar to beamformer 7B, and the means [[9']] g' here having ~~g₁' and g₂'~~ g'₁ and g'₂ as echo filters, whose coefficients may be influenced or set. The respective echo filter means g'_i (generally for i = 1, 2, [[..]] ...) are simple filters, which normally each have fewer coefficients than the number of coefficients in the corresponding echo canceller means g_i, at least in the case of reverberation cancellation, because then only the reverberation tail part of the simulated impulse response has to be copied by schematically shown copying means C1, C2 and C3 from the canceller means g_i into the filter means g_i'. Similarly the filter characteristics or coefficient values of the filters f_1, f_2, b_1, and b_2 are copied into the respective filters f_1', f_2', b_1', and b_2' of ~~beamfilter~~ beamformer 7'B. Now an auxiliary input signal is applied on input terminal 12 of the mirrored circuit arrangement ~~g', 7'~~ (i.e, the mirrored circuit arrangement including echo filter g' and circuit arrangement 7') such that the adaptive filter w' in circuit arrangement 7' which is coupled to output S' is capable of minimizing its output signal on output S' (i.e., wherein simulated audio distortion representative filter coefficient values are thus created by the minimizing of the output signal on output S' of adaptive filter w', the simulated audio distortion representative filter coefficient values representing correlation properties of reverberant tail parts of reverberation type audio distortion of a given sound field, and when copied to the filter arrangement, the simulated audio distortion representative filter coefficient values are for use by the filter arrangement in

suppressing reverberation type audio distortion in the given sound field). The auxiliary input signal on terminal 12 may for example be a stationary white noise signal or any other signal depending on the specific type of distortion to be cancelled. The thus simulated audio distortion representative filter coefficient values of the filter w' are copied into the filter coefficients of the filter w in filter arrangement 7 (i.e., shown in the upper portion of Fig. 1). These values represent the correlation properties of the reverberant tail part(s) of the sound field and can also be used for actually filtering the desired signal in filter arrangement 7 for suppressing reverberation therein, irrespective whether desired speech is output or not. In order to adapt or update the filter 7'B beamformer coefficients, only the reverberant behaviour of the room needs to taken into account. Thereto the desired audio source is not required, as any source in the room would do that job. As far long as the filter w receives its coefficients from the adaptive filter w', the filter w does not have to be adaptive.